REMARKS

Claims 1-50 were pending prior to the amendment. Claims 1-50 stand rejected. Claims 1, 10-13, 15, 17, 20, 22, 24, 33-34, 36-37 and 48 have been amended. Claim 14 has been cancelled. New claims 51-57 have been added. Reconsideration and allowance is respectfully requested.

Claim Rejections – 35 U.S.C. § 103

Claims 1, 2, 4, 6-9, 12, 24, 25, 27, 29-33, 35, 36, 41-45 and 47 have been rejected under 35 U.S.C. §103(a) as being unpatentable over Kung, et al. (U.S. Patent No. 6,775,267), hereafter Kung.

Claim 1 has been amended. Applicant claims changing an encoding algorithm at a telephone endpoint that is an originating source for a VoIP call in response to a user request and during the middle of the VoIP call. *See* figure 1 of the present specification. None of the references teach this feature alone or in combination.

Kung teaches a call center that controls how much network resources are provided for a call in response to a user request. Changing the amount of network resources can increase the size of "pipes" carrying the call. Based on the telephone meeting on November 16, 2005, Applicant believed this issue to be settled.

However, the Office Action indicated that, although Kung is expressly directed to changing how much network resources are provided to a call in response to a user, a relevant portion of Kung has not been previously discussed on the record. *See* page 19 of the Office Action dated February 23, 2006. The Office Action alleges that this un-discussed portion shows that "signaling from a user through the network can specify … codec selection at the central station 200." *Id.* A paragraph containing the portion cited is quoted below:

> An Ethernet interface with a RJ-45 connector may be used to connect the voice gateway 232 to the central router 210 (for example, Gigabit Switch or High Speed Router (HSR)). The multimedia gateway control protocol may be used as the interface between the voice gateway 232 and the call manager 218. For example, call control, signaling, and multi-media stream, real time protocol (RTP) connections, IP addresses, UDP ports, codec choice, etc., may be configured in any suitable manner such as by using a multimedia gateway control protocol. In exemplary embodiments, audio streams may be passed directly between customer premises equipment 102 using real time protocol connections over, for example, a user datagram protocol (UDP). Thus, the multimedia gateway control protocol may be utilized to request the voice gateway 232 to initiate, cancel, and/or otherwise modify connections in order to set up and tear

down RTP media streams. A similar procedure may also be utilized to request continuity tests and results. Kung, col. 14, line 56 to col. 15.

The above portion of Kung indicates that the central station 200 has the capability to use MGCP to control a codec selection and other variables at a local voice gateway 232. Clearly, the station 200 must be able to control its own gateways 232 for configuration, debugging and maintenance reasons.

However, the above section, and all other parts of Kung, do not disclose changing a codec in the *middle of a call in response to signaling from a caller over a network*. First, there is no disclosure for the station 200 to control the codec selection dynamically during the middle of a call. Moreover, the codec choice is listed with UDP port configuration and IP address, which are typically changed non-dynamically by the station 200, e.g. when configuring or booting a gateway. Second, there is no disclosure to change a codec *in response to* signaling from a user over a network.

In summary, Kung discloses changing network resource assignment to vary call priority, call path, etc. Applicant has now shown that even the recently cited portion of Kung does not disclose "signaling from a user through the network can specify ... codec selection at the central station 200" as was alleged in the Office Action. Thus, for this and other reasons, claim 1 should be allowed. Claims 2, 4, 6-9 and 12 are dependant and should also be allowed.

Claim 24 has been amended. Applicant claims code for updating the codec mid-call to correspond to the desired level of user perceived audio quality only when the current jitter amount is less than a predetermined jitter amount that is preset and represents an acceptable perceivable sound quality. This feature prevents a user from simply "cranking up the gain" and reducing overall congestion thus having the opposite intended result of reducing overall sound quality. *See* page 7, lines 25-28 of the present specification.

Even if Kung did teach controlling which encoding process is used to packetize VoIP call data in response to a user request (Kung does not teach this feature), Kung fails to prevent a user from "cranking up the gain" and saturating the network resources allocated for the call. In Kung, codec selection in a gateway 232 is controllable by a central station 200. If Kung did disclose a user requesting higher fidelity codecs (which it does not), the station 200 would apparently blindly control the gateway 232 by selecting a higher fidelity codec. When network congestion is present, the blind selection would lead to greater congestion producing a drop in overall call quality despite the higher fidelity codec.

In contrast, claim 24 only updates the codec selection mid-call only when the current jitter amount is less than a predetermined jitter amount that is preset according to an acceptable perceivable sound quality. Also, in addition to all the missing elements described above, a prima facie case of obviousness has still not been established for the reasons Applicant previously described on the record. Thus, for these and other reasons, claim 24 should be allowed. Claims 25, 27, 29-33 and 35 are dependant and should also be allowed. Claim 36 has been amended and should be allowed for at least similar reasons as claim 24. Claims 35, 36, 41-45 and 47 are dependant and should also be allowed.

Claims 3 and 26 have been rejected under 35 U.S.C. §103(a) as being unpatentable over Kung in view of Karagiannis (U.S. Patent Publication No. 2002/0015395).

Claims 3 and 26 are dependant and should be allowed for at least the same reason as their base claims.

Claims 5, 10, 11, 13-15, 17-23, 28, 37, 39, 48 and 50 have been rejected under 35 U.S.C. §103(a) as being unpatentable over Kung, in view of Havens (U.S. Patent No. 6,735,175).

Claims 5, 10 and 11 and 28 are dependant and should be allowed for at least the same reason as their base claims.

Claim 13 has been amended. The alleged combination fails to disclose or suggest system-defined codec selection when the user-defined selection does not provide the "best chance" of improving sound quality given the current congestion measurements. *See* the present specification page 8, lines 27 and page 9, lines 1-2. Thus, claim 13 should be allowed. Claims 14-15 and 17-23 are dependant and should also be allowed.

Claim 37 has been amended and should be allowed for at least similar reasons as claims 1 and 24. Furthermore, none of the references disclose dynamically varying packet payload size in response to a user request indicating delays associated with the call. Thus, claim 37 should be allowed. Claim 39 is dependant and should also be allowed.

Claim 48 has been amended. Support for the amendment may be found in the present specification, page 9, lines 26-28 and page 10, lines 1-18. The alleged combination does not suggest packetizing a remaining portion of the call into second packets having a second packet payload size that is different than the first packet payload size in response to the one or more DTMF tones. This feature allows a user of a PSTN phone to adjust a call to account for delays associated with the call.

Kung discloses a system that provides the ability to change quality of service. Kung does not even suggest changing packet payload size, much less changing packet payload size at a gateway in response to DTMF tones. Moreover, Kung does not suggest that the call manager 218 controls the voice gateway 232 using DTMF tones to change packet size in the middle of a call. See col. 14, lines 56-67 with reference to FIG. 3.

Havens teaches that codec may be changed to change call fidelity. However, Havens does not suggest changing packet payload size in response to DTMF tones to address delays associated with the call. Thus, Havens is silent on whether a user can address delays associated with a call instead of just fidelity of the call. A call with near perfect sound fidelity and substantial delays is less desirable than a call with good sound fidelity and insubstantial delays.

In contrast, claim 48 includes packetizing a remaining portion of the call into second packets having a second packet payload size that is different than the first packet payload size in response to the one or more DTMF tones. This feature allows a user of a PSTN phone to dynamically vary packet size at a gateway to obviate delays associated with the call during the call. Thus, claim 48 should be allowed. Claim 50 is dependant and should also be allowed.

Claims 16, 38 and 40 have been rejected under 35 U.S.C. §103(a) as being unpatentable over Kung, in view of Havens as applied to claims 15 and 37, and further in view of Karagiannis.

Claims 16, 38 and 40 are dependant and should be allowed for at least the same reasons as their base claims.

Claims 34 and 46 have been rejected under 35 U.S.C. §103(a) as being unpatentable over Kung in view of Havens as applied to claim 11, and further in view of Rosenberg, et al. (U.S. Patent No. 6,141,788).

Claims 34 and 46 are dependant and should be allowed for at least the same reason as their base claims.

Claim 49 has been rejected under 35 U.S.C. §103(a) as being unpatentable over Kung, in view of Kato (U.S. Patent No. 5,844,918).

No amendments have been made to claim 49. Neither reference teaches dynamically varying packet payload length … to correspond with the requested level of user perceived sound quality.

For example, in Kato error detection is achieved by appending headers as shown in figs. 5a-5d. There is no suggestion to dynamically change packet payload length to correspond to a user request. Moreover, none of the references on the record teach this feature.

In contrast, claim 49 includes the element of dynamically varying packet payload length to correspond with the requested level of perceived sound quality. This feature, for example, allows the system to reduce perceived delays in a VoIP call. See the present specification, page 4, lines 20-26. Thus, claim 49 should be allowed.

New Claims

New claims 51-54 have been added. Support for the new claims may be found in the present specification, page 4, lines 20-26 and Application 09/181,947, which is referred to in the present specification. Application 09/181,947 is not prior art at least for the purposes of obviousness. *See* 35 USC 103(c).

New claims 55-57 have been added. Support for the new claims may be found in the present specification, page 11, lines 22-25.

CONCLUSION

For the foregoing reasons, reconsideration and allowance of claims 1-57 of the application is solicited. The Examiner is encouraged to telephone the undersigned at (503) 222-3613 if it appears that an interview would be helpful in advancing the case.

Respectfully submitted,

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